CANCELLATION OF THE TONAL COMPONENT

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Abstract—**This study addresses the development of algorithms for the removal of tonal components from acoustic signals through active narrowband noise reduction methods. The core technique involves introducing a compensating sinusoid to the original signal, with parameters derived from a thorough analysis. However, the parameter estimation process introduces a processing delay.**

Two methods for integrating the compensating signal are proposed. The first involves adding the signal to the same frame used for parameter estimation, suitable for scenarios with acceptable processing time delays, such as in audio recordings. The second method introduces a synthesized compensating signal in the frame immediately following parameter estimation, catering to applications requiring minimal processing time, like narrowband noise reduction in signal processors or amplifiers. We have implemented the first method.

Illustrative examples demonstrate the efficacy of both methods, particularly in scenarios with tonal signals exhibiting slowly varying parameters. Overall, this research contributes valuable insights and practical approaches to tonal component elimination in acoustic signals, promising effective results in diverse applications, from audio recordings to electro-acoustic devices.

Keywords—Tonal Components, acoustics, Cancellation, signal processing.

I. INTRODUCTION

The problem of removing tonal components from signals has many different applications. It can be used to describe both the lowering of the tonal noise level and the elimination of tonal character distortions from pre-recorded signals in the electro-acoustic track. Moreover, tonal component removal can be helpful for measurement and diagnostic purposes since the effective removal of a component validates the accuracy of the parameter values that were estimated for it.

Additionally, eliminating the tonal components beforehand facilitates the examination of the noise components.Generally speaking, tonal disturbances are noticeably louder than broadband disturbances.Depending on how audible the tone is, the correction is added to the measured sound level . This approaches the perceived sound level more closely in the final adjusted result. This validates the requirement for tonal interference to be treated differently.

A simple tone, represented by a sinusoidal function, is the most basic type of tonal signal. Its spectrum consists of only one frequency. The majority of real signals are characterized by some change in time, hence it is rare to meet such a signal in practice. Numerous musical instruments are known to produce sounds that are composed of both fundamental and harmonic tones.

For example, a specific mathematical model of vibration based on sinusoidal functions has been employed in string instruments where the vibrator is a vibrating string. Rapid video cameras have been used to do thorough analyses.When the captured photos are analyzed, it becomes clear that there is considerable variability in addition to the events the model describes. These signals do not quite match their theoretical model in practice. The noises produced by air movement are another illustration.

Through the use of numerical analysis, models that produce sounds made up of tones and their harmonics under certain conditions have been confirmed . Discrete frequency values were not seen there, it was mentioned. The sinusoidal signal model is rarely used in practice, despite being very illustrative, user-friendly, and an excellent place to start for many investigations.

Using the example above, we cannot claim that our signal is made up of only one frequency (along with harmonics) when the string vibration frequency varies somewhat over time. The idea that the tonal component's frequency is contained in an interval surrounding a discrete frequency value would be a suitable definition. We can classify airflow-generated tones as narrowband noise in this instance. Because of this, signals must be systematized because tonality is distinct.

The method's theoretical underpinnings were first outlined in 1975, with an emphasis on the method's possible application for the removal of tonal interference for both and video signals either about electrocardiography or antenna technology [2].Solutions derived from this approach include attenuation [3], active transformer noise reduction [4] in active attenuation headphones [5], or attenuation of certain tone frequencies within a three-dimensional container [6], which is applicable in automobile interiors.

One significant drawback of these techniques is the requirement to enter data regarding the frequency of the removed tonal component. This data could have come from the processing of the original signal, from non-acoustic sensors, or prior knowledge [7].

The employed technique is predicated on the assessment of tonal component values, the calculation of their parameters (amplitude, frequency, and beginning phase), and the creation (synthesis) of suitable signals that let the lowering of the level of particular components. Still, downloading for synthesis adds a delay and takes some time.

II. BLOCK DIAGRAM

The algorithm we implemented uses the Interpolation of amplitude, phase, and frequency, zero padding, FFT, We applied a window function so that we can analyze the input signals section-wise.The Fast Fourier Transform (FFT), whose length expressed in samples equals a power of two, is the foundation of the algorithm used to estimate the values of tone component parameters. The frequency resolution and signal length that the FFT analyses are generally trade-offs.

One should select a long analysis window to achieve good signal resolution. On the other hand, the analysis must be done for the fewest available signal samples when dealing with time-varying signals. Such a procedure results in increased estimate errors and decreased frequency resolution of the FFT. Zero-padding and the interpolation of tonal component parameters improved the algorithm's efficiency [8][9].

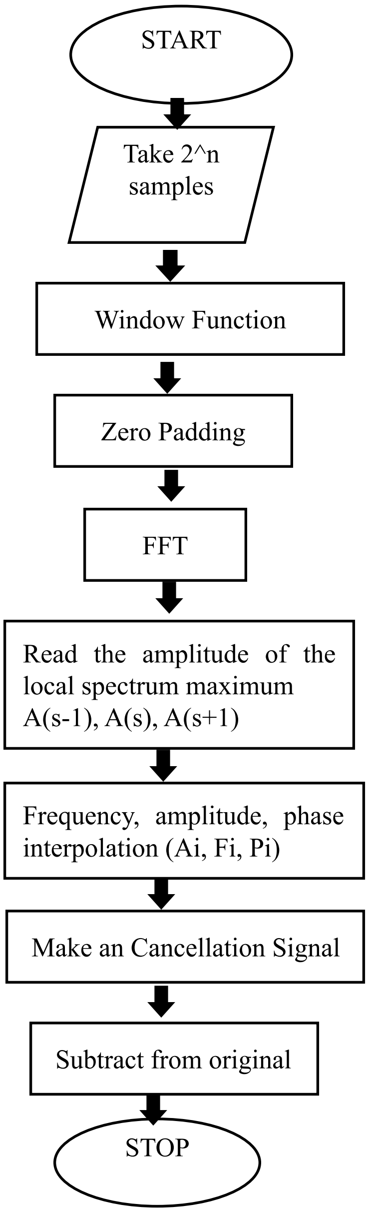

Figure 1. Block diagram of the algorithm for estimating parameter values using zero padding and FFT

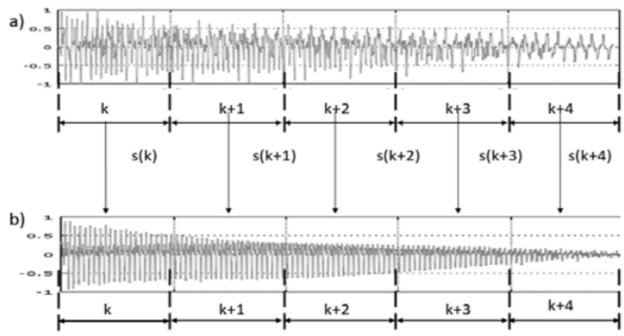

Figure 2. Adding a synthetic compensating signal (s = compensatory signal - k + i = subsequent signal frames) to the current signal frame.
a) the initial signal; b) the signal after the tonal component was eliminated.

III.ELIMINATION IN THE CURRENT FRAME

The estimation of the parameter values was carried out by appending the input signal to the same frame. When the processing time delay is tolerable, this technique can be applied to digital audio streams. The processing of audio recordings or the evaluation of the precision with which component parameter values in measurement systems are estimated are two examples of this. It can also be used in audio signal processing systems, such as digital radio receivers, where a particular buffer is present.

The process of adding a synthetic compensating signal, represented by a basic tone, to the original signal's frame—from which the parameters of the tonal component meant for synthesis were derived—is schematically depicted in Fig. 2.

Frames k, k+1, and k+2 are used to process the signal... The procedure of estimating the parameter values of the chosen tonal component of the original signal is done in each frame. Subsequently, a compensating signal is generated with parameters derived from the obtained parameter values, using formulas (1a), (1b), and (1c).

$$A_s = A_d \tag{1a}$$

$$f_s = f_d \tag{1b}$$

$$\varphi_s = \varphi_d \tag{1c}$$

where:
A_d, f_d, φ_d - amplitude, frequency, and starting phase of the original tonal component derived with the estimation procedure;
A_s, f_s, φ_s - amplitude, frequency, and initial phase of the synthesized compensatory signal.

IV. SIMULATION

An method with an FFT of length 2n was simulated as part of the investigation of the elimination performance in the ensuing phase-compensated frame.

The removal of this component could result in a decline in reduction efficiency over time as the signal parameters fluctuate. Consequently, the elimination process was carried out for a frequency-tuned signal that had a one-second duration, an increasing frequency (from 445 Hz to 561 Hz), and a decreasing amplitude (from 0.7 to 0.3). N=128 samples is the smallest length of the FFT for which parameter value estimation is possible. Parameter interpolation, four-fold zero padding, and fifty percent overlap were employed in the estimate procedure.

Figure 4 shows the initial fragments of the signal for elimination in the current frame and the next frame for an FFT length of N = 128 samples.

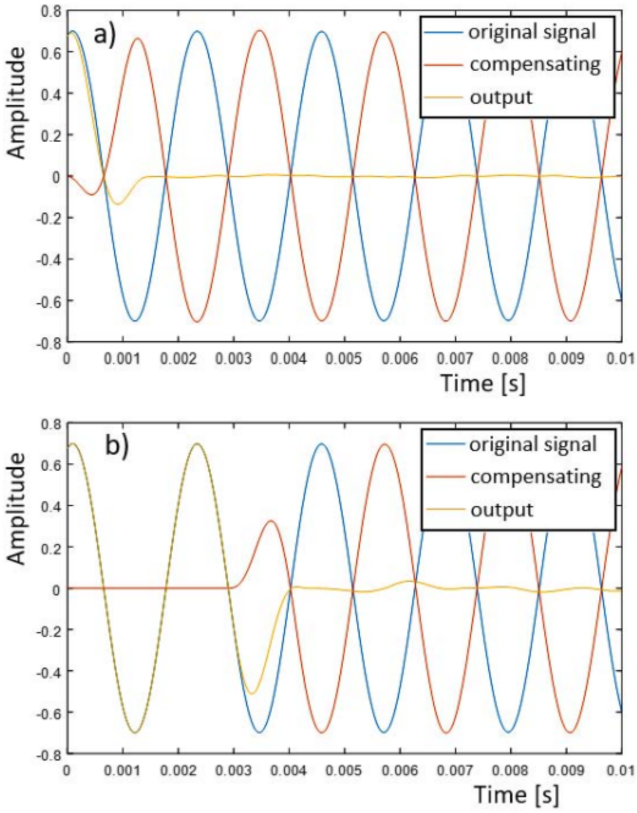

Figure 9. Fragments of the initial signals for
a) elimination in the current frame (without compensation) and
b) elimination in the following frame (with compensation)

The efficiency of the reduction declines as more samples are used in the algorithm's execution. This results from the signal's fluctuation inside a single window, the variation increases with the length of the window. The greater mismatch between the compensating signal and the original signal is the outcome of this. This strategy can effectively reduce its level if the adjusted tonal component's characteristics vary within a suitably limited frame and do not vary excessively.

V.TESTING THE ALGORITHMS

The phenomenon known as acoustic feedback usually shows up as a rising level that is restricted to a single frequency. It's a tone signal with amplitude increasing. To eliminate auditory feedback, notch filters are the most useful technique. However, because some information is removed using this method, the signal quality is reduced.

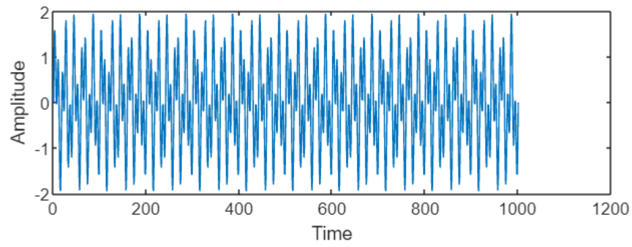

Figure 3. Input signal

Figure 3 shows the input signal we assumed, the results are observed from the below figures.

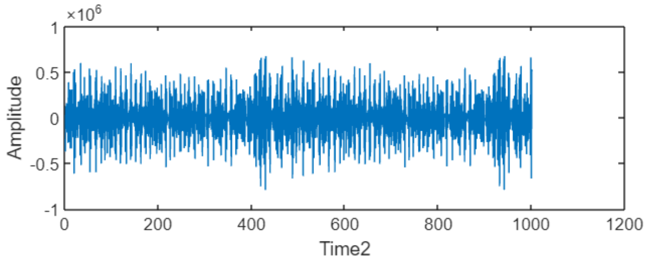

Figure 4. Output signal after cancellation using the hamming window function

Figure 4 shows the output signal after removing the tonal components we got after interpolation, we used the hamming window function.

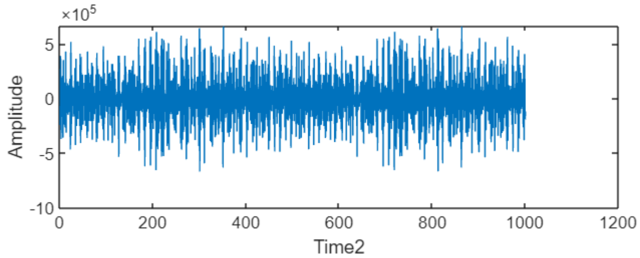

Figure 5. Output signal after cancellation using Bartlett window function

Figure 5 shows the output signal after removing the tonal components we got after interpolation, we used Bartlett window function.

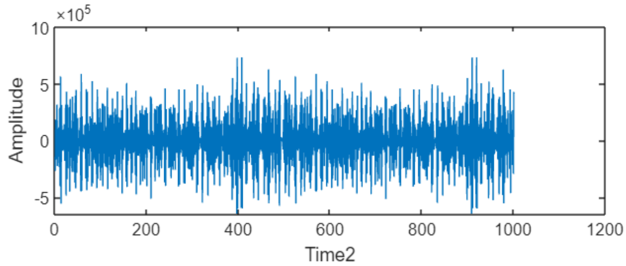

Figure 6. Output signal after cancellation using the Gausswin window function

Figure 6 shows the output signal after removing the tonal components we got after interpolation, we used the Gausswin window function.

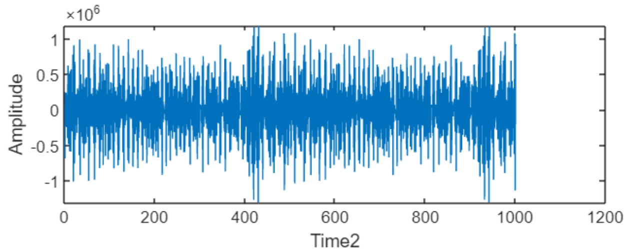

Figure 7. Output signal after cancellation using the Kaiser window function

Figure 7 shows the output signal after removing the tonal components we got after interpolation, we used the Kaiser window function.

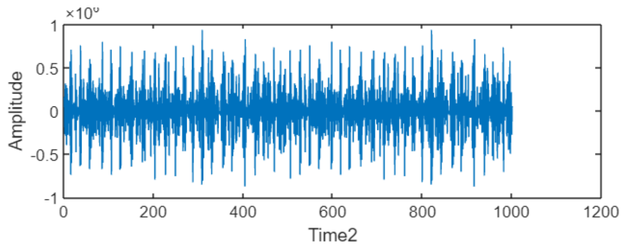

Figure 8. Output signal after cancellation using the Tukeywin window function

Figure 8 shows the output signal after removing the tonal components we got after interpolation, we used the Tukeywin window function.

VI. CONCLUSIONS

The method for eliminating tonal components using active narrowband noise reduction algorithms and a synthesized compensatory signal is presented in the paper. The procedure of estimating the parameters of the original signal's tonal component yielded the sinusoidal signal that serves as the compensating signal. A technique was used to add a compensating signal to the original signal, with parameters predetermined based on the signal frame prior, in order to offset the delay caused by the acquisition of samples required for the estimate procedure.
The elimination is still effective even though this method reduces the impact of tone component level reduction.

This version of the method provides for a more efficient reduction of distortion levels, but it also includes a delay due to the acquisition of signal samples. When employed in audio consoles or loudspeaker processors (that is, as a component of an electro-acoustic track), elimination should be utilized in the subsequent frame in conjunction with phase adjustment. In this manner, there are no delays caused by the algorithm. The degradation of the reduction efficiency for signals with time-varying properties is a drawback of this kind of approach.
More work is intended to implement more effective methods for tonal component parameter estimation and to integrate the suggested approach with active sound field noise reduction systems.

VII. REFERENCES

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